

WHAT IS CLAIMED IS:

1 1. A method of manipulating a digitized sound signal transferred over
2 a packet switched network in the form of data packets, the method being operable in a
3 communication system at a receiver end arranged to decode received sound data packets
4 into sound signal frames to be played back, the method including the step of:

5 manipulating the length of received signal frames by performing time
6 expansion or time compression of one or more signal frames at time varying intervals and
7 with time varying lengths of the expansion or the compression, said intervals and said
8 lengths being determined so as to maintain a continuous flow of signal samples to be
9 played back.

1 2. The method of claim 1, wherein each of said time varying lengths
2 is dependent upon a signal fitting criteria with respect to the signal characteristics of the
3 digitized sound signal part to be manipulated.

1 3. The method of claim 1, wherein a resolution of the length
2 manipulation is a fraction of the time between two samples of said digitized sound signal,
3 thereby enabling an improved signal fitting quality when performing said time expansion
4 or said time compression.

1 4. The method of claim 1, wherein the receiver end includes a jitter
2 buffer, and including the steps of

3 storing received data packets to be decoded into signal frames, and

4 monitoring the jitter buffer to initiate the manipulating step when the
5 timing of the jitter buffer needs to be recovered.

1 5. The method of claim 4, wherein the time expansion of one or more
2 signal frames is performed for a trailing part of a currently played signal frame if the
3 monitoring of the jitter buffer indicates near or actual buffer underflow, and the
4 manipulating step including repeated time expansions to restore the jitter buffer to its
5 normal working condition.

1 6. The method of claim 5, wherein said time expansion of said trailing
2 part will constitute a substitution frame if the monitoring of the jitter buffer indicates that
3 a next signal frame, which under normal conditions should follow the currently played
4 signal frame, is not available or deemed not to have been received in due time, thereby
5 providing a lost frame substitution for said next signal frame, after which the time
6 expanded currently played signal frame is merged with a received future signal frame, the
7 length of the time expansion, and thus the length of the substitution frame, being chosen
8 in such way that a smooth transition to said future signal frame can be made.

1 7. The method of claim 6, wherein said time expansion includes time
2 expanding a heading part of said future signal frame before merging the two frames,
3 thereby improving the lost frame substitution.

1 8. The method of claim 4, including initiating time compression by
2 real-time statistics from the jitter buffer when two consecutive data packets are available
3 in the jitter buffer, wherein a measure on a smooth transition between two consecutive
4 signal frames, when merging the two signal frames, controls the length of a resulting
5 compressed signal frame.

1 9. The method of claim 8, wherein said compressed signal frame is
2 merged with yet another consecutive signal frame in the same manner as said merging of
3 said two frames.

1 10. The method of claim 8, wherein said merging of two signal frames
2 involves merging two signal segments, a trailing segment of one frame with a heading
3 segment of the other frame, by overlap-add, wherein a time-shift of the frame with the
4 heading segment is employed for optimizing the matching of the overlapping part of the
5 two segments.

1 11. The method of claim 10, wherein said time-shift of the frame with
2 the heading segment has a resolution of a fraction of a time between two samples.

1 12. The method of claim 11, wherein said heading segment is
2 multiplied with a suitable gain to further optimize the matching with said trailing segment

3 at the overlapping part, after which a smooth transition back to unity gain is performed in
4 order to avoid sound signal discontinuities.

1 13. The method of claim 1, wherein use is made of an oscillator model
2 for extracting signal segments used when manipulating the lengths of said received signal
3 frames, the oscillator model including a codebook in which vectors of samples forms
4 different states, or entries, in the codebook, the codebook storing a corresponding signal
5 segment for each state.

1 14. The method of claim 13, wherein said time expansion of a signal
2 frame is performed by matching a true state of a trailing part of such signal frame with
3 said states in said codebook, and reading out a signal segment from said codebook that
4 corresponds to the state having been matched with said true state.

1 15. The method of claim 13, wherein said signal segments of said
2 codebook have variable lengths, each signal segment forming a trailing part of a signal
3 frame, thereby enabling continuous transition from the time expanded signal frame to a
4 consecutive signal frame.

1 16. The method of claim 13 wherein time delays between said states in
2 said codebook are incremental delays with a resolution of a fraction of a time between
3 two samples.

1 17. The method of claim 14, wherein the states and the corresponding
2 segments of said codebook are scaled in order to improve the matching with said true
3 state.

1 18. The method of claim 14, wherein merging of said true state is
2 performed with the matching state of said codebook.

1 19. The method of claim 14, wherein said time expansion additionally
2 involves performing the corresponding operations with respect to a heading part of a
3 signal frame being consecutive to the time expanded signal frame.

1 20. The method of claim 1, wherein said signal frame, which length is
2 to be manipulated, is either a sound signal frame resulting from a complete decoding

3 operation of a data packet, or an intermediate time-domain signal frame resulting from a
4 partial decoding operation of a data packet.

1 21. The method of claim 1, including the step of using an oscillator
2 model, which oscillator model includes a codebook in which vectors of samples of a
3 received digitized sound signal forms different states, or entries, in the codebook, the
4 codebook storing a corresponding signal segment for each state.

1 22. The method of claim 1, including storing, in a program storage
2 device, a sequence of instructions for a processor unit for performing the method of claim
3 1.

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1 23. Apparatus for receiving a digitized sound signal from a packet
2 switched network, the arrangement including:
3 a memory element for storing a computer program and vectors of samples
4 of a received digitized sound signal together with corresponding signal segments; and
5 a processor unit for executing the computer program to decode the
6 received sound signal and produce therefrom sound signal frames to be played back by
7 manipulating the length of received signal frames by performing time expansion or time
8 compression of one or more signal frames at time varying intervals and with time varying
9 lengths of the expansion or the compression, said intervals and said lengths being
10 determined so as to maintain a continuous flow of signal samples to be played back.

1 24. An article of manufacture including a computer memory wherein is
2 located a computer program for causing digitized sound signal transferred over a packet
3 switched network in the form of data packets to be received by a receiver unit of a
4 communication system and to decode received sound data packets into sound signal
5 frames to be played back by manipulating the length of received signal frames by
6 performing time expansion or time compression of one or more signal frames at time
7 varying intervals and with time varying lengths of the expansion or the compression, said
8 intervals and said lengths being determined so as to maintain a continuous flow of signal
9 samples to be played back.

1 25. A receiver unit for receiving digitized sound in the form of data
2 packets over a packet switched network, the receiver including a processing element
3 having a memory wherein is located a computer program for causing said receiver unit to
4 decode received sound data packets into sound signal frames to be played back by
5 manipulating the length of received signal frames by performing time expansion or time
6 compression of one or more signal frames at time varying intervals and with time varying
7 lengths of the expansion or the compression, said intervals and said lengths being
8 determined so as to maintain a continuous flow of signal samples to be played back.

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